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Pulse-code modulation for high-quality sound-signal distribution: a simulation of instantaneous companding

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RESEARCH DEPARTMENT

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A.H. Jones, B.Sc.

D. Maurice
Head of Research Department

(EL-27)

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PULSE-CODE MODULATION FOR HIGH-QUALITY SOUND-SIGNAL DISTRIBUTION : A SIMULATION OF INSTANTANEOUS COMPANDING

SUMMARY

This report describes a series of subjective tests made to discover whether instantaneous companding could profitably be applied to a pulse-code-modulation system designed for high quality sound signals. It was concluded that the benefits derived from this type of companding would be too small to justify the additional instrumental complexity involved.

1. INTRODUCTION

The development of transmission systems and of digital techniques has reached a stage where it is both desirable and feasible to apply pulse code modulation (p.c.m.) to high-quality sound signals. The principles of p.c.m., together with a description of a possible system for high quality sound distribution, were dealt with in an earlier report.[1] That report also considered various methods of improving the signal-to-noise ratio at the output of such a system by carrying out certain processes on the analogue or the digital signals.

One of the techniques referred to was high-frequency pre- and de-emphasis and it was suggested that a still further improvement in output signal-to-noise ratio might be possible if the gain reduction within a protective limiter following the pre-emphasis circuit could be compensated at the receiving end of the system.

A later report[2] described a syllabic compandor arrangement which made use of these concepts and was capable of achieving a signal-to-noise ratio improvement of 15 dB. A practical embodiment of this system[3] gave a 13 dB improvement, the difference being due to instrumental deficiencies.

Other possible methods of improving a p.c.m. system suggested in the early report[1] involved a non-linear relationship between the instantaneous values of the sound signal and the corresponding digital codes. If a linear relationship exists between the analogue and digital quantities, the effect of quantizing the signal is to introduce a noise component, which is more audible at low signal levels than at high. This tendency may however be offset by redistributing the quantizing levels so that they are closer together in the region of signal excursion occupied by low-level signals than in the regions additionally used by high-level signals. This technique, the subject of the present investigation, will be referred to as 'instantaneous companding', and is an important feature of the p.c.m. systems used for telephony.[4]

A number of methods of achieving the redistribution of quantizing levels have been suggested, the non-linearity being introduced on one side or the other of the analogue-digital interface.

Instantaneous compandors operating on the analogue signals have been made using the non-linear characteristics of diodes.[5] Such devices are suitable for telephone systems, where intelligibility is more important than quality, but they are unsuitable for distribution systems handling high-quality signals because it is difficult to match the characteristics of the non-linear circuits adequately at the sending and receiving terminals.[6]

Other methods of instantaneous companding include an adjustment of the gain and d.c. level of the signal applied to the analogue side of the analogue-to-digital converter (automatic ranging coding[1]), adjustment of the reference signal with which it is compared (this may be applied to successive-approximation a.d.cs [7] or to counter a.d.cs and purely digital operations such as altering the count rate in a counter a.d.c. The last of these techniques can be applied with complete accuracy; there is no difficulty involved in matching the characteristics of the two terminals.

Although instantaneous companding removes some of the impairments caused by the use of a small number of digits per word, however, it can, if not carefully applied, introduce a new type of impairment. Clearly if one chooses a quantizing level density in the centre of the signal excursion range which is adequate to prevent quantizing distortion of the low level signals that occupy this region, there is then a limit to the amount by which one can reduce the density outside this region without the quality of high-level signals becoming noticeably impaired. Therefore before applying such measures to a p.c.m. system, one needs to know what degree of instantaneous companding can be tolerated by the listener. It was for this purpose that the subjective tests described in this report were carried out.

The investigation made use of a simulation technique, which produced the effect of a p.c.m. system combined with instantaneous companding perfectly instrumented. Pre- and de-emphasis was included in some of the tests to see if a greater improvement could be obtained from instantaneous companding by this means.

The investigation was not exhaustive or rigorous, the results of a preliminary series of tests being such as to discourage a fuller examination of this type of companding; sufficient work was however done to enable some firm conclusions to be drawn.

2. METHOD OF SIMULATION

The simulation procedure adopted for this investigation made use of the fact that a p.c.m. system which is supplied with an undistorted input signal but whose available quantizing levels are non-linearly disposed in relation to the range of signal excursion has a performance identical to that of a p.c.m. system whose quantizing levels are linearly disposed *in relation to signal excursion* but which is supplied with a non-linearly distorted input signal and has a complementary non-linear network connected at its output.

An 11-digit linear a.d.c./d.a.c. pair was used, which had a sampling rate of 31·25 kHz and an audio bandwidth of 14 kHz. (The accuracy of the a.d.c. and d.a.c. were such that when operated normally a signal-to-quantizing-noise ratio[1] within 1 dB of the theoretical value was obtained.) The spacing between quantizing levels appropriate to 11 digits was always preserved at the centre of the range of signal excursion, but was made variable outside this range. This was achieved by operating the a.d.c. in the normal fashion, but preventing the d.a.c. from responding to one or more of the least significant digits when the instantaneous magnitude of the input signal was large.

It is not possible to predict with certainty the results that would have been obtained if the density in the central region had been appropriate to, say, a 12-digit system, but it seems quite likely that the general conclusions drawn would still be valid. It would not be safe however to apply them to systems in which the audio-frequency bandwidth is very different from that used in the present investigation.

Fig. 1 is a block diagram showing the arrangement used. The signal magnitudes at which changes in the quantizing level density were made were decided by two comparators whose reference voltages could be adjusted as required. Each comparator was fed with a full-wave rectified version of the input signal. The output of the comparator was therefore a signal indicating when the instantaneous magnitude of the audio signal lay outside a central range whose extremities had been fixed by adjustment of the reference voltage and which was symmetrically disposed with respect to the mean signal level. The comparator output was delayed by the time taken for the corresponding p.c.m. signal to reach the d.a.c., and was then fed to the d.a.c. to cause certain of the digits to be inhibited. The inhibit signal caused ones to be written into the d.a.c. input register positions occupied by those digits, regardless of the actual composition of the incoming words. The effect of doing this is indicated in Fig. 2. This shows at (*a*) a representation of some of the quantizing levels appropriate to the 11-digit system, together with the corresponding state of the three least significant digits. Now if, for example, the last two digits are set to ones, only those levels marked with an asterisk will be employed; if an intermediate level is indicated by the digital code, the next higher asterisked level will be substituted. Fig. 2(*b*) shows the decoded signals that result from applying to this system an input voltage which increases steadily by one quantizing step during each sampling interval. If, as in the left-hand half of Fig. 2, all of the quantizing levels are available for use, the output filter removes the steps in the sample-held signal waveform to give a true replica of the input step.

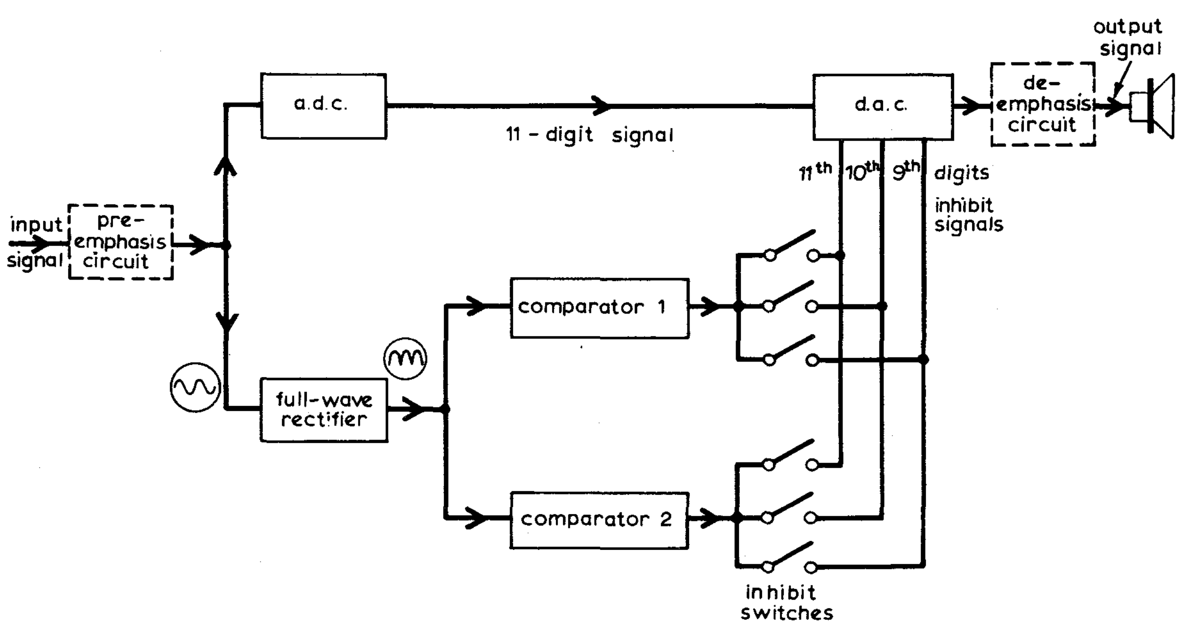

Fig. 1 - Block diagram of equipment

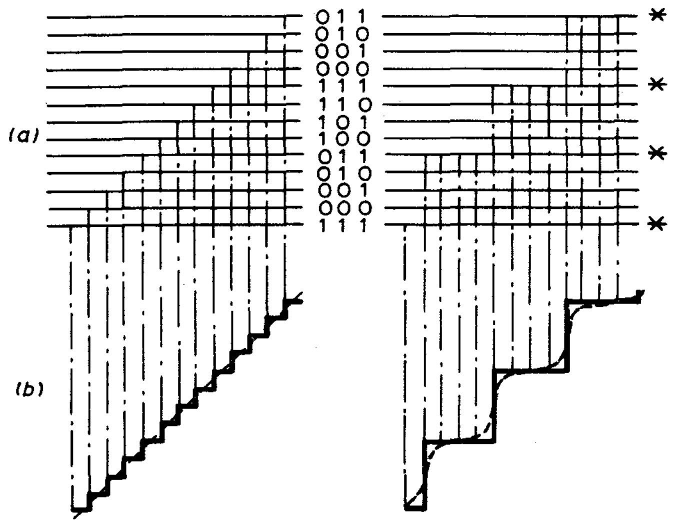

Fig. 2 - Effect of digit inhibition on signal quantization
——— output signal before filtering
- - - - output signal after filtering

In general the input voltage would not, as in this simple example, rise by an integral number of quantizing levels in one sampling interval and there would be a small discrepancy between the output and the input voltage waveforms. This discrepancy is brought about by the quantizing errors associated with 11-digit coding.

Much greater discrepancies are produced, however, when some quantizing levels are not used. The right-hand half of Fig. 2 shows how the exclusion of three quarters of the 11-digit quantizing levels by the inhibition of the two least significant digits gives rise to an output of the form associated with 9-digit coding.*

Fig. 1 shows the positions at which pre- and de-emphasis units were inserted during some of the tests. The pre-emphasis characteristic was that recommended for carrier systems by the CCITT[8] and used in modified form in the sound-in-vision system.[3] The curve was effectively set down, however, by a reduction of gain, so that the gain at 1 kHz was not 0 dB but −3 dB. This prevented perceptible overloading of the p.c.m. system by programme material having high-level high-frequency content, and in the absence of instantaneous companding gave an effective reduction of 4 dB in quantizing noise. Thus if 11-digit linear coding, giving 2048 quantizing levels, were satisfactory without pre- and de-emphasis, 1300 levels would be sufficient when the above pre- and de-emphasis was used.

* It can be shown that this method of coarse quantization simulation by inhibition of least significant digits produces a slight rise in the average instantaneous signal magnitudes in those parts of the waveform where it is used. This introduces unwanted but low-level even-order harmonic distortion. The observers' descriptions of the distortions they heard, however, confirmed suspicions that the effects of coarse quantization were very much more noticeable than those of low-order harmonic distortion, and that the latter could therefore be neglected.

3. COMPRESSION CURVES SIMULATED

The arrangement described above is equivalent, as the beginning of Section 2 suggested, to one in which the same relationship between instantaneous signal magnitude and digital code is secured by operations on the analogue signal at the input and output of a p.c.m. system in which the coding and decoding are completely linear. It is therefore convenient to describe the simulated signal processing in terms of the transfer characteristic of an equivalent compressor circuit connected at the input of a linear p.c.m. system.

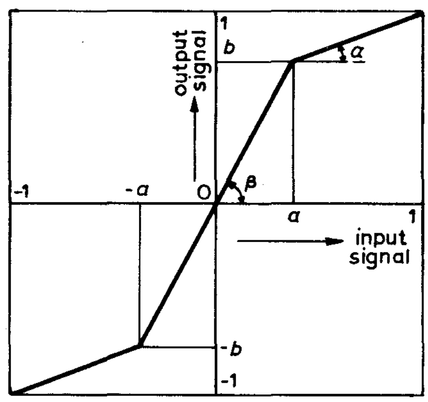

Fig. 3 - Form of an equivalent compression characteristic

Fig. 3 shows the type of equivalent compression characteristic obtained when one comparator, giving one transition point on each side of the zero crossing is used in the simulation. The input and output signals are normalized to have a peak value of ±1.

Suppose the changeover from fine to coarse quantization occurs when the input signal is at magnitude a. Suppose too that r of the binary digits are inhibited for magnitudes in excess of a. The ratio of quantizing level densities above and below a is then given by

$$1 : 2^r$$

In the equivalent compressor characteristic shown in Fig. 3, the knee occurs at input signal = a, the ratio of slopes above and below the knee being given by

$$\frac{\tan\alpha}{\tan\beta} = \frac{1}{2^r}$$

$$\text{i.e. } \frac{(1-b)}{(1-a)} \cdot \frac{a}{b} = \frac{1}{2^r}$$

$$\text{therefore } b = \frac{a.2^r}{1+a(2^r-1)}$$

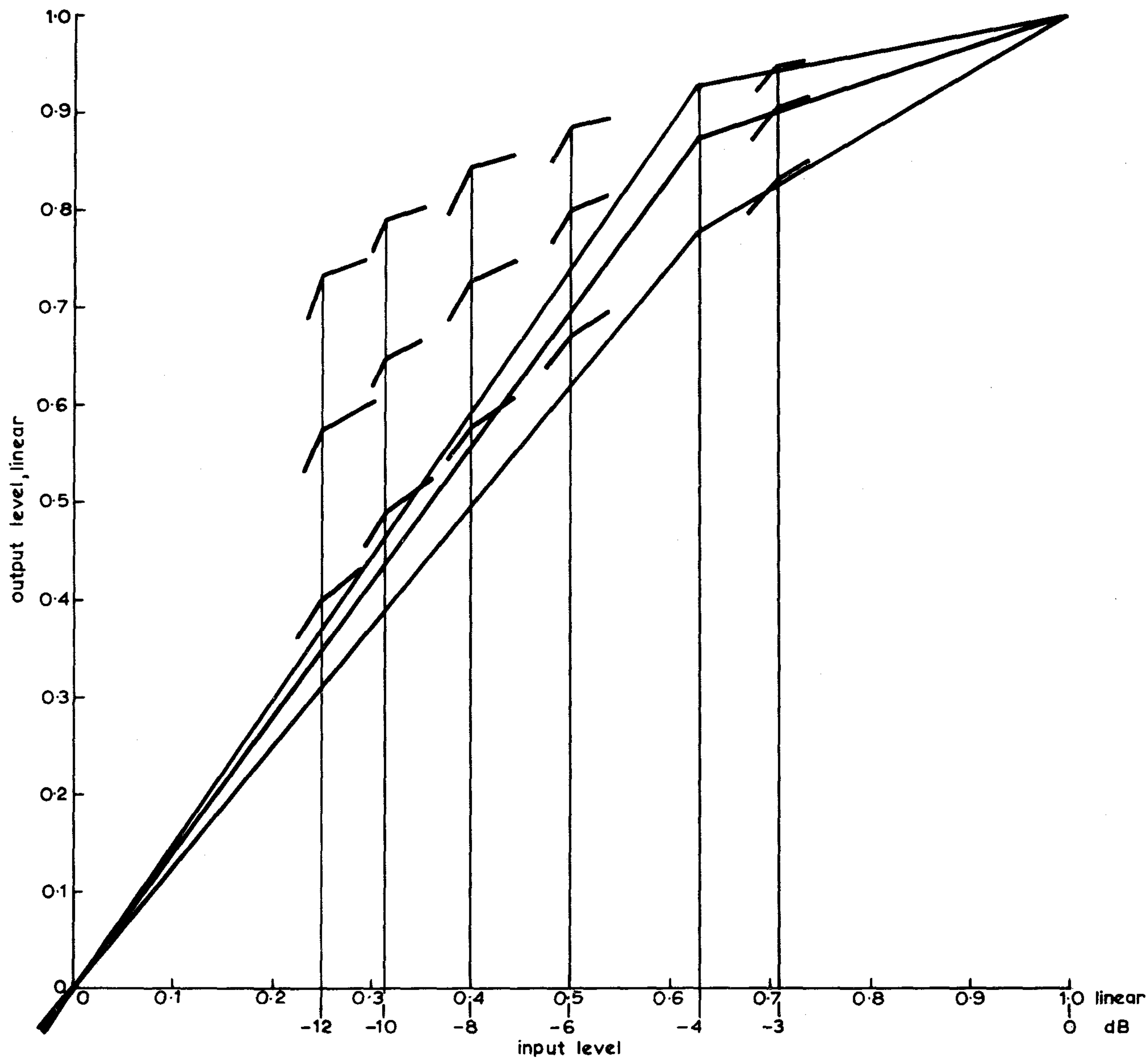

Fig. 4 - Compression characteristics equivalent to single-transition test conditions (positive half only shown)

Thus if a and r are given, b can be calculated and the equivalent compressor characteristic plotted. Note that although a (given by the comparator reference voltage) could be made to have any desired value, the method of simulation required that r should be an integer. Nevertheless a wide range of characteristics was simulated. The transition points were set at either −3, −4, −6, −8, −10 or −12 dB relative to peak signal level, and at each of these positions either the 11th, or the 10th and the 11th, or the 9th, 10th and 11th digits were switched. The corresponding compression characteristics are shown in Fig. 4. The three curves associated with the −4 dB transition point are shown in full, and the points at which other slope changes were simulated are indicated.

In order, however, to assess the practical value of instantaneous companding, one needs to know how many quantizing levels may be saved when it is used. The total number of quantizing levels available under the conditions illustrated in Fig. 4 is shown in Fig. 5; curves (a) relate to this system without pre- and de-emphasis, while curves (b) take into account the reduction in quantizing noise obtained when pre- and de-emphasis is used. These curves are based on the assumption that in the absence of instantaneous companding or pre- and de-emphasis 2048-level quantizing noise would be acceptable. It is further assumed (See Section 2) that if in the absence of these artifices a different number of levels were required the results of the present investigation could be scaled accordingly. The 13 dB improvement obtained in practice with a suitable syllabic compandor[3] is shown for comparison at (c) as equivalent to a reduction from 2048 to 457 levels.

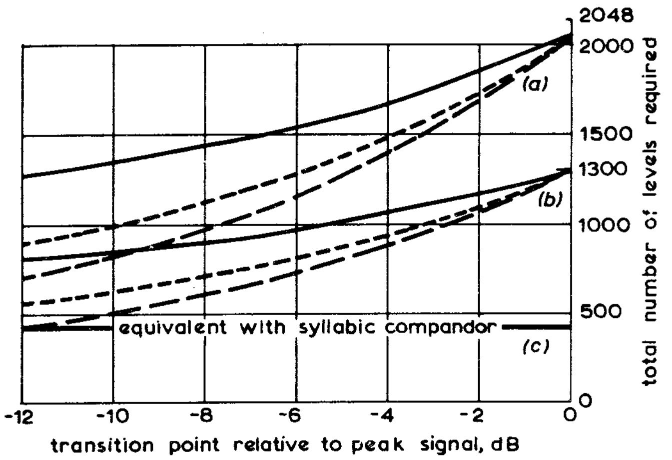

Fig. 5 - Graph showing number of levels assumed to be required when instantaneous companding of the form shown in Fig. 3 is used

(*a*) without pre- and de-emphasis
(*b*) equivalent with pre- and de-emphasis
——— least significant digit removed above transition point
- - - - - least 2 significant digits removed above transition point
— — — least 3 significant digits removed above transition point

A somewhat more complicated pair of equations can be calculated to give the equivalent compression characteristics obtained when two transitions were introduced by using both comparators. The characteristics simulated are shown in Fig. 6.

4. THE SUBJECTIVE TESTS

Preliminary listening tests revealed that the distortion introduced by instantaneous companding, when apparent, had the appearance of high-order harmonic distortion; this form of distortion is particularly apparent when piano music is reproduced. A tape recorded excerpt of piano music was therefore used as programme material for the tests, and its level was set to fully modulate the p.c.m. system.

A number of observers, most of whom were experienced in the assessment of quality, took part in the tests. A high-quality monitoring loudspeaker was used; the listening level was adjusted to the satisfaction of the observers and then left constant throughout the tests.

An extreme example of the type of distortion to be expected was demonstrated. The piano excerpt was then played four times in succession, the four presentations being labelled 'A', 'B', 'A' and 'B' respectively. Either one of conditions A or B involved linear 11-digit p.c.m. The other included the simulation of instantaneous companding. The observers were asked to decide whether A was inferior in quality to B, or vice-versa. They were allowed to vote 'don't know' if they could not detect any difference.

This method of presentation was chosen to give the observers the best opportunity of detecting impairments

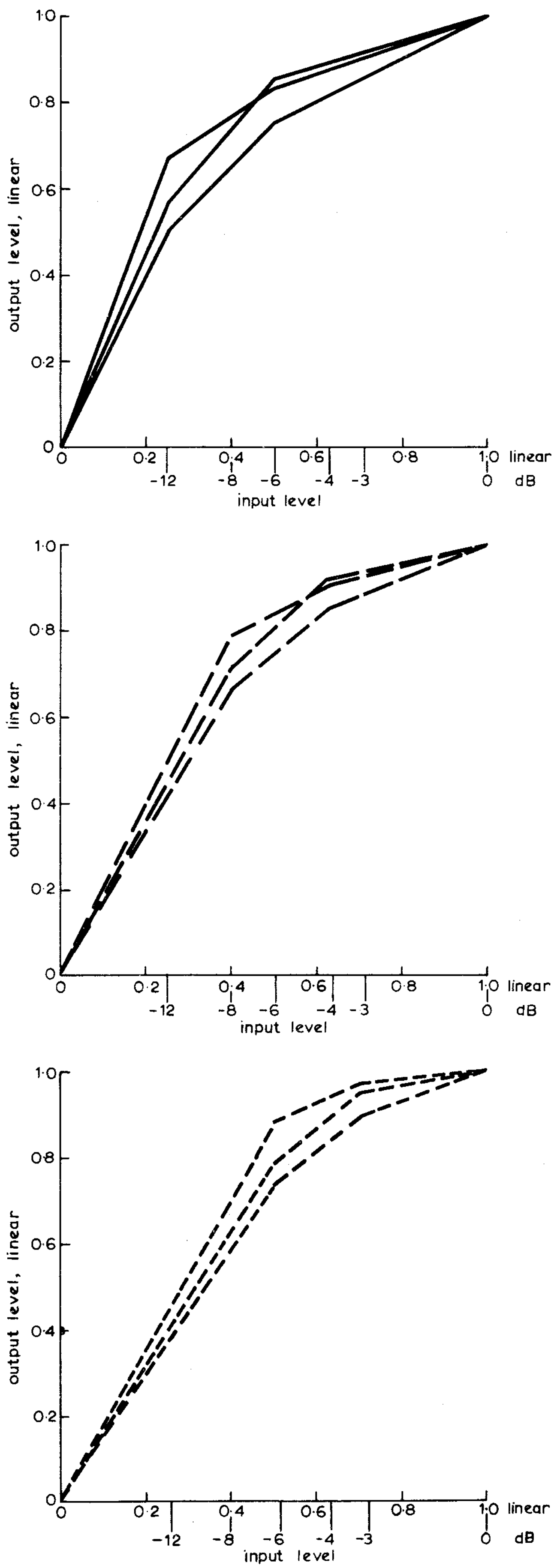

Fig. 6 - Compression characteristics equivalent to double-transition test conditions

introduced by instantaneous companding while discouraging them from guessing the answer if the difference was imperceptible.

There were 8 listening sessions; a total of 11 observers took part in the tests in groups of four or five.

5. RESULTS

The total numbers of 'right', 'wrong' and 'don't know' answers obtained for each simulation were counted. The three figures, x, y and z respectively, were then processed as follows. It was assumed that those whose vote was 'wrong' did not detect a difference between the conditions presented and therefore guessed (though they may not have realized it). It is then only fair to assume that of the x 'right' answers an equal number, y, were similarly the result of guesswork. The proportion of observers who really did detect the impairment due to companding of the type simulated is then given by

$$\frac{x-y}{x+y+z}$$

This figure is then the probability that such impairment would be detected.

The results of the tests involving a single pair of transition points are given in Fig. 7. Curves (*a*) without pre-emphasis result from a total of 256 observations, and curves (*b*), with pre-emphasis, from 120 observations.

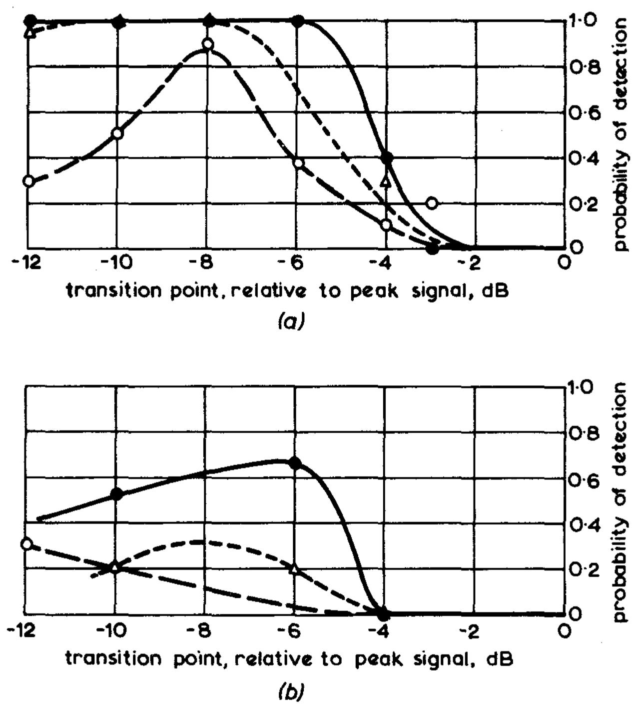

Fig. 7 - Results of tests involving a single pair of transition points

(*a*) without pre-emphasis (*b*) with pre-emphasis

—○— least significant digit switched

--△-- least two significant digits switched

—●— least three significant digits switched

The results shown in Fig. 7 are replotted in Fig. 8 using information from Fig. 5(*a*). Fig. 8 shows the total number of levels used and hence the saving in levels that can be achieved using instantaneous companding against the probability that the departure from uniform coding will be noticed. Superimposed on Fig. 8 are crosses giving the results obtained from tests in which the two comparators (Fig. 1) were used to simulate characteristics having two pairs of transition points (Fig. 6). (These tests did not include pre- and de-emphasis.)

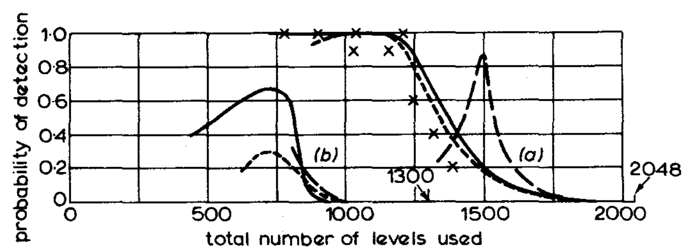

Fig. 8 - Graphs obtained by combining Figs. 5(a) and 7(a), with points relating to double pair of transition points superimposed

(*a*) without pre-emphasis (*b*) with pre-emphasis

—·—·— least significant digit switched

- - - - - least 2 significant digits switched

——— least 3 significant digits switched

x x x results obtained using two pairs of transition points (without pre- and de-emphasis)

6. DISCUSSION

The curves plotted in Fig. 8 are based on a limited number of observations and were obtained using only one (albeit a quite critical) musical excerpt as programme material. Thus the precise form of the curves given in Fig. 8 is probably not of great significance. Nevertheless some general conclusions can be drawn:

(i) The positions of the crosses in Fig. 8, corresponding to the use of two pairs of transition points, suggest that there is some advantage to be gained from making the compression characteristic smooth rather than as a series of chords. Since the crosses are not far removed from the other curves shown in set (*a*), however, it appears that this advantage is not great and that not much more would have been gained by further smoothing the equivalent characteristics.

(ii) Pre- and de-emphasis is advantageous with as well as without instantaneous companding. If one concentrates on the region of interest, where the distortions are just becoming apparent and observes the position of curves (*a*) relative to the point marking 2048 levels (11 digits), comparing that with curves (*b*) relative to 1300 levels (equivalent to 11 digits when pre- and de-emphasis is heard), it becomes apparent that the percentage saving in quantizing levels gained from pre- and de-emphasis is very slightly greater when instantaneous companding is used than when it is not used.

(iii) Although some saving in quantizing levels (less than the equivalent of 1 bit) can be gained by using instantaneous companding, the degree of improvement achieved is unlikely to justify the instrumental complication involved. Once pre- and de-emphasis, together with the necessary protective limiter, have been introduced, the additional complexity required for instantaneous companding is probably at least as great as that required for syllabic companding,[2] while the latter is much more effective. It was because of this conclusion that the investigation was not carried further.

(iv) It is of some interest that there is a tendency for the curves in Figs. 7 and 8 to rise to a peak when the transition point is lowered and then to fall again. This suggests that the signal distortion introduced by having a transition in the characteristic (variously described by the observers as 'fizziness', 'raggedness', 'crushing noises' – forms commonly used to describe the effects of high order harmonic distortion) was more noticeable than the increased quantizing noise caused by maintaining the reduced quantizing level density throughout the whole range of signal excursion. The distortion observed might not, however, have been so obvious if the listening level had been substantially reduced. It is just possible, therefore, that if a sufficient number of levels were chosen to give acceptable quantizing noise over the normal range of programme levels, instantaneous companding restricted to the region immediately on either side of zero signal magnitude might overcome a form of quantizing distortion[9] sometimes apparent when the signal is slowly faded to a very low level. The apparatus available for the tests could not, however, be conveniently adapted to test this possibility.

7. CONCLUSIONS

The tests carried out have indicated that little worthwhile improvement to a high quality p.c.m. sound system can be obtained by incorporating instantaneous companding; the improvement is certainly much less than that given by a syllabic compandor. Instantaneous companding should not therefore be considered as a useful means of reducing the quantizing noise associated with a p.c.m. audio system with a restricted number of digits. It may possibly have some value, however, if applied in the region occupied by signals of a very low level as a means of overcoming the quantizing distortion that becomes apparent at these levels.

8. REFERENCES

1. Pulse-code modulation for high-quality sound signal distribution : appraisal of system requirements. BBC Research Department Report No. EL-10, Serial No. 1967/50.

2. A frequency-dependent compandor system for high-quality sound signal distribution. BBC Research Department Report No. EL-21, Serial No. 1968/52.

3. ROUT, E.R. *and* JONES, A.H. The use of pulse code modulation for point-to-point music transmission. *Radio & Elect. Engrs.*, **37**, No. 1, April 1969, pp. 199 – 207.

4. BOLTON, L.J. *and* BENNETT, G.H. Design features and application of the British Post Office 24-channel pulse-code-modulation system. *P.O.E.E.J.*, July 1968, Vol. 61, Part 2, pp. 95 – 101.

5. MANN, H., STRAUBE, H.M., *and* VILLARS, C.P. A companded coder system for an experimental p.c.m. terminal. *Bell System Tech. J.*, January 1962, pp. 173 – 226.

6. Pulse-code modulation for high-quality sound signal distribution : instantaneous compandors using diodes as non-linear elements. BBC Research Department Report No. EL-15, Serial No. 1968/7.

7. TERRY, J.B. Encoding, decoding and multiplexing in a speech p.c.m. system. IEE Colloquium on p.c.m., Colloquium Digest No. 1968/7, Paper 4, pp. 1 – 7.

8. CCITT. Red Book, Vol. III, p. 273.

9. Pulse-code modulation for high-quality sound signal distribution : quantizing distortion at very low signal levels. BBC Research Department report in preparation.

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